The MaximalSound Algorithm

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Date: July 2010

This document provides a complete overview of the MaximalSound algorithm. It's organised according the processing order.

1. Analyze
2. Harmonic Enhancement
3. Crossover
4. De-Expander
5. Limiter
6. Video Links

1 Analyze

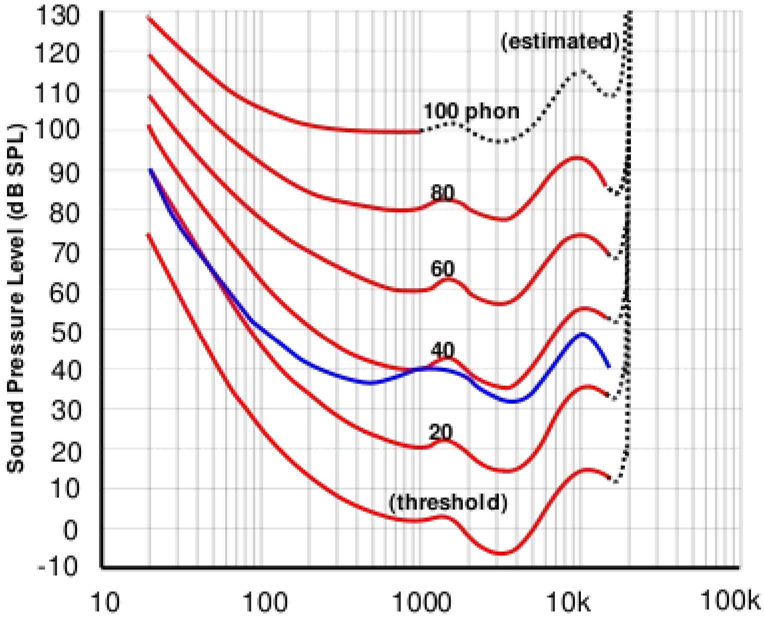
Because the MaximalSound service is a fully automated pre-mastering process, the first step of the treatment is an analyze of the whole file. This allows to optimize settings of the downstream processing. Unfortunately, this also prevents to run the algorithm in real time.

2 Harmonic Enhancement

The harmonic enhancer, this old invention of the Aphex company, has the advantage on an equalizer to not apply a fixed amplification on the top of the audio spectrum. This technique allows to naturally clarify the sound without fatiguing constant high frequencies boost. In the MaximalSound algorithm the even and odd harmonics are generated separately depending on whether the attack or the body of the note are processed. This adds some vacuum tube character to your sound.

3 Crossover

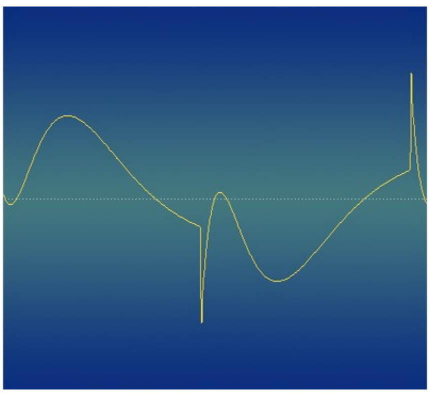
A 32 bands crossover filter splits the signal to be processed by 32 de-expanders before being rebuilt in broad-band by summing all the bands individually processed. This produces a dynamic equalizer where each part of the audio spectrum is magnified in the low levels according to a psycho acoustic model standard.



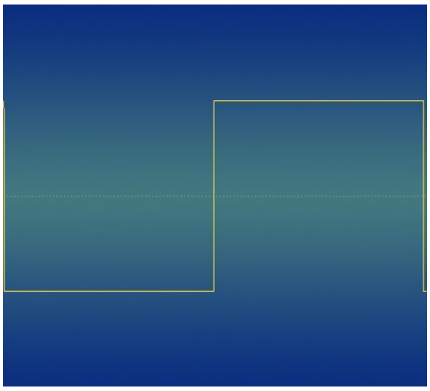
Equal-loudness contours (red) (from ISO 226:2003 revision)
Original ISO standard shown (blue) for 40-phons

Because this model was established using pure and constant tones, an adaptation has been done to make it suitable for the variable complex timbers of music. This particular adjustment is purely empirical. It only relies on thousands hours of attentive listening.

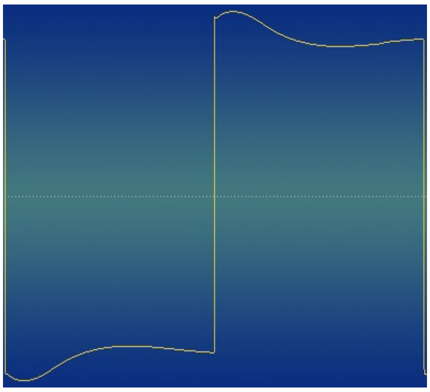
Despite an IIR (infinite impulse response) filter solution has been chosen, the maximal possible phase error is reduced to a 5° margin on the complete processing. This provides a perfect reconstruction of a square wave, by preserving the time relation between frequencies on the whole audio spectrum. The FIR (finite impulse response) filter topology has the theorical advantage to feature both linear phase and constant group delay across the whole audio bandwidth. But it has been rejected after listening tests, due to harsh artifacts when the de-expander treatment was applied.



The previous picture displays a square signal reconstructed after passing thru a 3 poles IIR crossover. The peak and RMS values of the original square signal are drastically modified. The human hearing isn't very sensitive to these changes, but all downstream processors will have to deal with this changes.



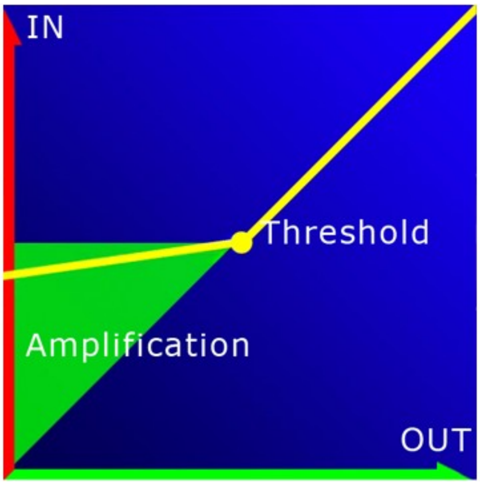
The picture above displays a square signal reconstructed after passing thru the MaximalSound 32 bands crossover (no other treatment applied). The downstream processors won't be affected by any peak and RMS values modification. This point is very important when dealing with dynamics.



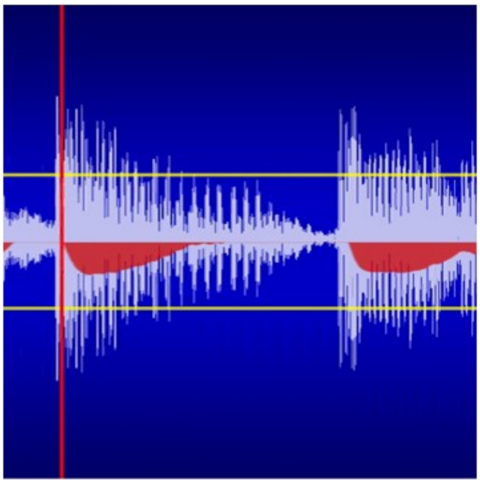
This is a picture of a full scale square signal passing thru the complete MaximalSound processing (harmonic enhancer, crossover, multi-band de-expander, limiter).

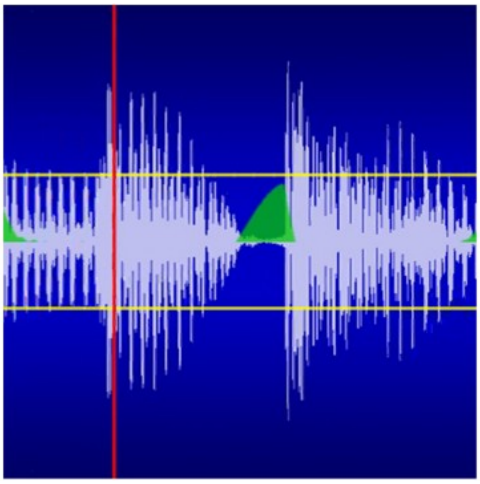
4 De-Expander

A de-expander can also be called reverse-expander (negative ratio) or upward compressor. A de-expander amplifies the signal below a threshold in contrast to an expander which attenuates the signal below this threshold. This choice is fundamentally different than techniques related to compression. The signal is amplified when it is weak and not planed when it becomes too high (above threshold).



The difference may seem subtle, but it is significant when the attack time of treatment is taken into account . A compressor will act always too late (attack time) on the peak values, forcing the limiter to affect the transients which are so valuable to the perception of space. The de-expander just amplifies signal after the attack time, avoiding all unnecessary limiting.





This technique can't be compared with parallel compression since two different parts of the signal are processed. Note that no “look-ahead” is applied. This technique is very effective to prevent any threshold violation and produce a very smooth sound, but it usually kills the sound impact, and it may become very unnatural for ears.

5 Limiter

Unlike the analog domain, digital doesn't support any overload, even briefly. The limiter has to handle situations where the summation of all processed bands produces a signal exceeding the allowed maximum value (0 dB FS) in the digital domain. The limiter is an essential part of treatment since at this stage the signal is full-band and any defect may lead to tonal unbalance by favoring or attenuating certain parts of the audio spectrum. In limiters the attack time is zero or negative (look ahead) to prevent any violation of the limit value. The art is to find an optimum release time, in order to preserve the frequency content and the perception of the dynamics of the original signal regardless of the complexity of this signal.

6 Video Links

- MaximalSound At Work
- The Phase Matters
- Multiband De-expander
- Compressor vs De-expander
- Dynamic Processors by Transfer Curves
- Digital Overload
- All Limiters aren't Equal

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